MULTIMEDIA UNIVERSITY

STUDENT ID NO

MULTIMEDIA UNIVERSITY

FINAL EXAMINATION

TRIMESTER 1, 2017/2018

ETM7166 – DIGITAL SIGNAL PROCESSING SYSTEMS AND DESIGN IN TELECOMMUNICATIONS

(All sections / Groups)

24 OCTOBER 2017
8:00 PM – 11:00 PM
(3 Hours)

INSTRUCTIONS TO STUDENTS

1. This Question paper consists of 12 pages with 4 Questions only.
2. Attempt **ALL** questions. All questions carry equal marks and the distribution of the marks for each question is given.
3. Please print all your answers in the Answer Booklet provided.

Question 1

(a) Digital signal processing (DSP) algorithms have continued to find great use in increasingly wide application areas.

(i) Briefly describe any two advantages of DSP over analogue signal processing.

[2 marks]

(ii) With a block diagram, briefly explain the process of discrete-time digital processing of analogue signals.

[5 marks]

(iii) Briefly describe the options available to implement DSP systems for different applications in practice.

[3 marks]

(b) Consider a digital filter whose output $y[n]$ is related to the input $x[n]$ through the following relationship:

$$y[n]=\frac{1}{M}\sum_{k=0}^{M-1}x[n-k]$$

where M is a finite integer. Verify whether the system is causal and linear.

[5 marks]

(c) Figure Q1 shows the impulse response of a linear time-invariant (LTI) system, evaluate the system output for the input sequence $x[n]=3\delta[n]-2\delta[n-1]$.

[4 marks]

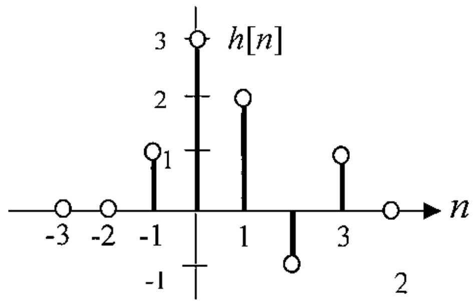

Figure Q1

(d) An LTI system is characterized by the system function:

$$H(z)=\frac{9-3.2z^{-1}-0.48z^{-2}}{1+0.3z^{-1}-0.88z^{-2}+0.24z^{-3}}=\frac{1}{1-0.5z^{-1}}+\frac{2}{1-0.4z^{-1}}-\frac{6}{1+1.2z^{-1}}$$

(i) Obtain the linear constant coefficient difference equation (LCCDE) of the LTI system.

[2 marks]

(ii) Plot the region of convergence (ROC) if the LTI system is stable.

[2 marks]

(iii) Plot the ROC if the LTI system is causal.

[2 marks]

Continued...

Question 2

(a) The N-point discrete Fourier transform (DFT) is defined by:

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j2\pi kn/N} \qquad \text{for } k = 0, 1, \ldots, N-1$$

(i) Quantify the computational complexity of the DFT in terms of the number of complex multiplications and additions required to obtain N DFT samples.

[3 marks]

(ii) Compute $X[0]$ and $X[1]$ for the following sequence using $N = 4$.

$$x[n] = \delta[n] + 2\delta[n-1] + 3\delta[n-3] + 4\delta[n-4].$$

[5 marks]

(b) Define the frequency response of an ideal digital filter and sketch the frequency responses of ideal digital lowpass, highpass, bandpass and bandstop filters, respectively.

[5 marks]

(c) A CD quality audio signal is sampled at a frequency of 44.1kHz. The signal has a useful content up until 15 kHz and is contaminated by noise from 16 kHz onwards. Design a digital finite impulse response (FIR) low-pass filter that can attenuate the noise by at least 50 dB without affecting the useful content by more than 1 dB.

(i) Analyze whether the pass-band or stop-band ripple requirement is more stringent.

[2 marks]

(ii) Based on (i), determine the particular fixed window that should be used in the filter design.

[2 marks]

(iii) Design the filter by obtaining the complete equation for the filter impulse response $h[n]$.

[4 marks]

(d) In the context of infinite impulse response (IIR) filter design using the bilinear transformation method, explain what is meant by frequency warping and briefly describe the practice to address this issue.

[4 marks]

Continued...

Question 3

(a) State 3 reasons why we need to consider many different structures when designing discrete-time systems.

[3 marks]

(b) Figure Q3 shows the input and output sequences of a particular multirate system.

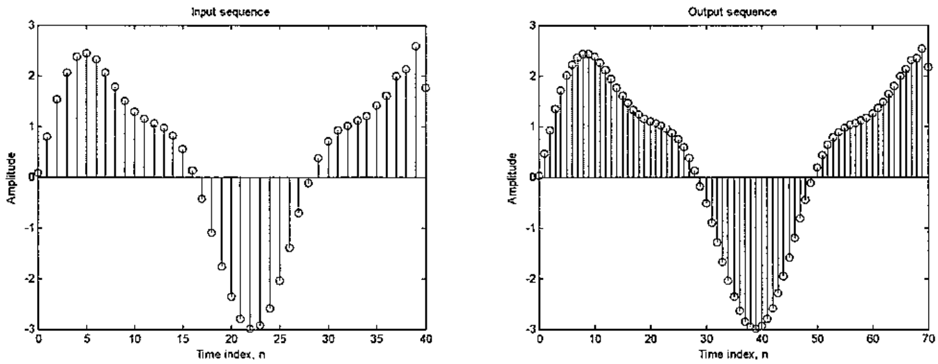

Figure Q3

(i) Determine the sampling rate conversion factor of the system.

[2 marks]

(ii) The above system can be implemented by cascading an interpolator with a decimator and a filter. Suggest the best implementation structure for the system.

[2 marks]

(iii) Explain the purpose of the filter in the structure, and specify its characteristics.

[3 marks]

(c) Consider a decimator for reduction of sampling rate from 30kHz to 2kHz. The specifications of the decimation filter $H(z)$ are assumed as follows:

Passband edge F_p = 800Hz,
Stopband edge F_s = 1000Hz,
Passband ripple δ_p = 0.02
Stopband ripple δ_s = 0.01

For a single stage implementation, the system requires 496,000 multiplications per second. It is suggested that the computational requirement of the decimator to be reduced by designing it as a two-stage structure, with $H(z) = F(z)G(z^5)$. Compute the computational requirements of the proposed two-stage structure and comment on the results.

[8 marks]

Continued...

(d) (i) The least mean square (LMS) algorithm is an iterative approach to the minimum mean square error (MMSE) performance in adaptive signal processing. As an alternative to the LMS algorithm, the recursive least squares (RLS) algorithm can also be employed to adapt the filter coefficients. Explain the advantages and disadvantages of these two algorithms.

[4 marks]

(ii) Describe how adaptive filtering can be used in system identification or system modelling applications.

[3 marks]

Question 4

(a) Audio coding techniques can be categorized into waveform, parametric and hybrid coders. Briefly describe the three coding techniques in terms of their underlying concept as well as their bit rate.

[4 marks]

(b) Pitch estimation is one of the most important processes in linear prediction coding. The estimated pitch period is used to generate the input signal for voiced segments. Briefly discuss two available methods in estimating the pitch period.

[3 marks]

(c) With appropriate diagrams, explain the problem of acoustic echoes in speakerphone applications and the use of adaptive filters to overcome the problem.

[6 marks]

(d) (i) Voicing detector is one of the many components in Linear Predictive Coding (LPC) encoder. Describe the use of energy, zero crossing rate, and prediction gain parameters as a voicing detector.

[6 marks]

(ii) To be more reliable, several voicing parameters are used to make a joint decision. Figure Q4 shows a speech waveform together with its energy, zero crossing, and prediction gain; all these parameters are calculated using a frame length of 180 samples. The parameters are assumed to be constant within the frame so that the plots have a staircase appearance. Discuss the voicing status of the signal for $n < 1200$, $1200 < n < 2100$, and $n > 2100$.

[6 marks]

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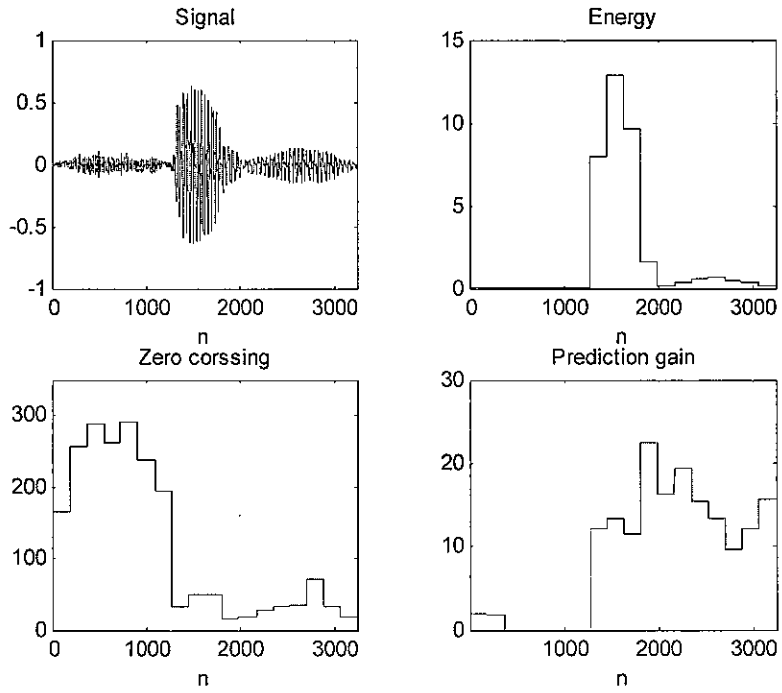

Figure Q4

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Appendix: Formula Sheet

The z-transform

Properties of the z-transform

Property	$x[n]$	$X(z)$	$\mathcal{R}_x$
Linearity	$ax[n] + by[n]$	$aX(z) + bY(z)$	$\mathcal{R}_x \cap \mathcal{R}_y$
Time shifting	$x[n-m]$	$z^{-m}X(z)$	$\mathcal{R}_x$
Time reversal	$x[-n]$	$X(z^{-1})$	$1/\mathcal{R}_x$
Convolution	$x[n] * y[n]$	$X(z)Y(z)$	$\mathcal{R}_x \cap \mathcal{R}_y$

Common z-transform pairs

$x[n]$	$X(z)$	$\mathcal{R}_x$
$\delta[n]$	1	$\forall z$
$\delta[n-n_0]$	z^{-n_0}	Possibly $\forall z$
$u[n]$	$\dfrac{1}{1-z^{-1}}$	$\lvert z\rvert > 1$
$-u[-n-1]$	$\dfrac{1}{1-z^{-1}}$	$\lvert z\rvert < 1$
$a^n u[n]$	$\dfrac{1}{1-az^{-1}}$	$\lvert z\rvert > \lvert a\rvert$
$-a^n u[-n-1]$	$\dfrac{1}{1-az^{-1}}$	$\lvert z\rvert < \lvert a\rvert$
$na^n u[n]$	$\dfrac{az^{-1}}{(1-az^{-1})^2}$	$\lvert z\rvert > \lvert a\rvert$
$-na^n u[-n-1]$	$\dfrac{az^{-1}}{(1-az^{-1})^2}$	$\lvert z\rvert < \lvert a\rvert$

Closed-form Expression for Some useful Series

$$\sum_{n=0}^{N-1} a^n = \frac{1-a^N}{1-a}$$

$$\sum_{n=0}^{N-1} na^n = \frac{(N-1)a^{N+1} - Na^N + a}{(1-a)^2}$$

$$\sum_{n=0}^{N-1} n = \frac{1}{2}N(N-1)$$

$$\sum_{n=0}^{\infty} a^n = \frac{1}{1-a} \qquad |a| < 1$$

$$\sum_{n=0}^{\infty} na^n = \frac{a}{(1-a)^2} \qquad |a| < 1$$

$$\sum_{n=0}^{N-1} n^2 = \frac{1}{6}N(N-1)(2N-1)$$

$$\sum_{n=N_1}^{N_2} a^n = \frac{a^{N_1-1} - a^{N_2}}{1-a}$$

Continued…

FIR Filter Design

Ideal bandpass

$$h[n] = \frac{w_2}{\pi}\,\mathrm{sinc}\left(\frac{w_2(n-M/2)}{\pi}\right) - \frac{w_1}{\pi}\,\mathrm{sinc}\left(\frac{w_1(n-M/2)}{\pi}\right),\quad n = 0, 1, \ldots, M$$

Fixed windows

Window	Window function
Rectangular	$w[n] = 1$
Hann	$w[n] = 0.5 - 0.5\cos(2\pi n/M)$
Hamming	$w[n] = 0.54 - 0.46\cos(2\pi n/M)$
Blackman	$w[n] = 0.42 - 0.5\cos(2\pi n/M) + 0.08\cos(4\pi n/M)$

Window	Passband ripple $20\log_{10}\delta_p$	Stopband attenuation $20\log_{10}\delta_s$	Transition width $\lvert w_p - w_s \rvert$
Rectangular	-13	-21	$1.8\pi/M$
Hann	-31	-44	$6.2\pi/M$
Hamming	-41	-53	$6.6\pi/M$
Blackman	-57	-74	$11\pi/M$

Kaiser window

$$w[n] = \frac{I_0\left(\beta(1-(n/\alpha-1)^2)^{0.5}\right)}{I_0(\beta)}$$

$$\beta = \begin{cases} 0.1102(A-8.7), & A > 50 \\ 0.5842(A-21)^{0.4} + 0.07886(A-21), & 21 \le A \le 50 \\ 0, & A < 21 \end{cases}$$

$$M = \begin{cases} (A-7.95)/(2.285\Delta w), & A \ge 21 \\ 5.655/\Delta w, & A < 21 \end{cases}$$

Optimal Filter Design (Parks-McClellan Algorithm) Estimated Filter Order

$$N = \frac{-20\log\sqrt{\delta_p\delta_s} - 13}{14.6\Delta f}$$

Continued...

CLASSIFICATION OF LINEAR-PHASE FIR SYSTEMS

	$h[n]$ symmetric: $h[n] = h[N-n]$	$h[n]$ antisymmetric: $h[n] = -h[N-n]$
N even	**Type I Linear Phase Filter** $H(e^{j\omega}) = e^{-jN\omega/2}\sum_{k=0}^{N/2} a[k]\cos(k\omega)$ $a[0] = h[N/2]$ $a[k] = 2h[(N/2)-k]$	**Type III Linear Phase Filter** $H(e^{j\omega}) = je^{-jN\omega/2}\sum_{k=1}^{N/2} c[k]\sin(k\omega)$ $c[k] = 2h[(N/2)-k]$
N odd	**Type II Linear Phase Filter** $H(e^{j\omega}) = e^{-jN\omega/2}\sum_{k=1}^{(N+1)/2} b[k]\cos((k-1/2)\omega)$ $b[k] = 2h[\frac{(N+1)}{2}-k]$	**Type IV Linear Phase Filter** $H(e^{j\omega}) = je^{-jN\omega/2}\sum_{k=1}^{(N+1)/2} d[k]\sin((k-1/2)\omega)$ $d[k] = 2h[\frac{(N+1)}{2}-k]$

Continued...

IIR Filter Design

Normalized Butterworth lowpass

N	a_1	a_2	a_3	a_4	a_5	a_6	a_7	a_8
1	1.000							
2	1.414	1.000						
3	2.000	2.000	1.000					
4	2.613	3.414	2.613	1.000				
5	3.236	5.236	5.236	3.236	1.000			
6	3.864	7.464	9.142	7.464	3.864	1.000		
7	4.494	10.10	14.59	14.59	10.10	4.494	1.000	
8	5.126	13.14	21.85	25.69	21.85	13.14	5.126	1.000

Filter order

$$d = \left(\frac{(1-\delta_p)^{-2} - 1}{\delta_s^{-2} - 1} \right)^{0.5}$$

$$k = \frac{\Omega_p}{\Omega_s}$$

Design	Filter order
Butterworth	$N \geq \dfrac{\log d}{\log k}$
Chebyshev I, II	$N \geq \dfrac{\cosh^{-1}(1/d)}{\cosh^{-1}(1/k)}$
Elliptic	$N \geq \dfrac{\log(16/d^2)}{\log(2/u)}$
	$u = \dfrac{1-(1-k^2)^{0.25}}{1+(1-k^2)^{0.25}}$

$$\cosh^{-1}(x) = \ln\left(x + \sqrt{x^2 - 1}\right)$$

Continued...

Frequency transformations

Target class	Transformation	Edge frequencies of target class
Highpass	$s \rightarrow \dfrac{\Omega_p \Omega'_p}{s}$	Ω'_p
Bandpass	$s \rightarrow \Omega_p \dfrac{s^2 + \Omega_u \Omega_l}{s(\Omega_u - \Omega_l)}$	Ω_l, Ω_u
Bandstop	$s \rightarrow \Omega_p \dfrac{s(\Omega_u - \Omega_l)}{s^2 + \Omega_u \Omega_l}$	Ω_l, Ω_u

Impulse invariance transformation

$$H_a(s) = \sum_{k=0}^{p-1} \frac{A_k}{s - s_k} \longrightarrow H(z) = \sum_{k=0}^{p-1} \frac{T_s A_k}{1 - e^{s_k T_s} z^{-1}}$$

Bilinear transformation

$$H(z) = H_a(s)\Big|_{s = \frac{2}{T_s}\frac{1-z^{-1}}{1+z^{-1}}}$$

$$\Omega = 2\tan(w/2)/T_s$$

Continued...

Discrete-time Fourier Analysis

The discrete Fourier transform (DFT)

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j2\pi kn/N}, \quad k = 0, 1, \ldots, N-1$$

Properties of the DFT

Property	$x[n]$	$X[k]$
Linearity	$A_1 x_1[n] + A_2 x_2[n]$	$A_1 X_1[k] + A_2 X_2[k]$
Time shifting	$x[\langle n - n_0 \rangle_N]$	$X[k] W_N^{kn_0}$
Frequency shifting	$x[n] W_N^{-k_0 n}$	$X[\langle k - k_0 \rangle_N]$
Time reversal	$x[\langle -n \rangle_N]$	$X[\langle -k \rangle_N]$
Conjugation	$x^*[n]$	$X^*[\langle -k \rangle_N]$
Convolution	$x[n] \circledast y[n]$	$X[k] Y[k]$
Modulation	$N x[n] y[n]$	$X[k] \circledast Y[k]$

End of Paper